IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant:	Jes Thyssen
Serial No.:	09/782,884
Filed:	February 13, 2001
Art Unit:	2655
Examiner:	Tran, Vincent V.
TITLE:	Speech Coding System with Input Signal Transformation

DECLARATION UNDER 37 C.F.R. § 1.131

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir/Madam:

I, Jes Thyssen, declare as follows:

1. I am the inventor of the subject matter described and claimed in the above-referenced United States Patent Application Serial No. 09/782,884, filed February 13, 2001, entitled "Speech Coding System with Input Signal Transformation."

2. I declare that I conceived invention of the subject matter of the above-referenced application in the United States, as defined by its pending claims 1-22, prior to July 14, 2000.

3. To evidence conception of invention of the subject matter of the above-referenced application in the United States, attached hereto, please find a copy of the submission I drafted on behalf of my then employer, Conexant Systems, Inc., to the ITU-T, entitled "Detailed description of the 4 kbit/s eX-CELP Algorithm Submitted by Conexant Systems for the ITU-T Qualification Phase," dated January 5, 2000.

4. I declare that Section 3.1.1.1 of the enclosed submission to the ITU-T, entitled "Silence Enhancement", evidences my conception of invention of the subject matter of the above-referenced application in the United States prior to July 14, 2000.

5. I declare that the present invention, as defined by claims 1-22 pending in the above-referenced application, was reduced to practice at Conexant Systems, Inc. in the United States at 4311 Jamboree Road, Newport Beach, California 92660, using due diligence after conception of invention of the subject matter of the above-referenced application.

6. I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine of imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the above-referenced patent application or any patent issuing thereon.

2-10-2004
Date

Jes Thyssen
Jes Thyssen

CONEXANT

Technical Report:

Detailed description of the 4 kbit/s eX-CELP Algorithm Submitted by Conexant Systems for the ITU-T Qualification Phase.

Version 3.0, January 5th 2000

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Contents

3 Detailed Description of the Algorithm

The detailed description of the eX-CELP algorithm is presented in this section. The purpose of the section is to provide insight to the algorithm and describe the fundamentals of the algorithm to an extent that allows engineers familiar with the field to understand the algorithm with support of the accompanying C-code and easily work on the algorithm. The C-code delivered with the description takes precedence in case of discrepancies. For very low level information on the algorithm please consult with the C-code.

3.1 Encoder Algorithm

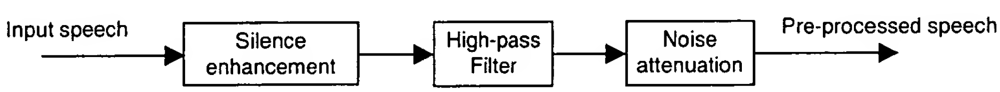
The encoder of the eX-CELP algorithm is described in this section. The functionality of each of the main processing functions of the encoder will be described in the following.

The block diagram of the encoder is divided into four figures, Figure 1, Figure 3, Figure 4, and **Error! Reference source not found.**. Figure 1 and Figure 3 illustrate the frame based processing functions that are common to both modes (Mode 0 and 1) of the algorithm. The pre-processing stages, that conditions the speech signal prior to encoding, are shown in Figure 1, and the common frame based encoding is presented in Figure 3. The subsequent processing functions dedicated to Mode 0 and 1, respectively, are depicted in Figure 4 and **Error! Reference source not found.**, respectively.

3.1.1 Pre-processing of Speech

Figure 1 depicts the pre-processing of the speech signal prior to the actual speech encoding. This includes silence enhancement, high-pass filtering, and background noise attenuation.

Figure 1: Pre-processing of speech.



3.1.1.1 Silence Enhancement

After reading and buffering the speech samples for a given frame the segment is analyzed in order to detect if the frame is pure silence, i.e. only "silence noise" is present. The function adaptively tracks the minimum resolution and levels of the signal around zero. According to this information the function adaptively detects on a frame basis whether the current frame is silence and the component is purely "silence-noise". If it is detected as "silence noise" the function ramps the signal to the zero-level of the signal. Otherwise the signal is not modified. The zero-level of the signal depends on the (unknown) processing prior to the speech coding algorithm. For A-law this value is 8, while for μ-law and 16 bit linear PCM it is 0. The zero-level of the signal is tracked adaptively by the function. It should be noted that the function in general only modifies the signal if the sample values for the given frame are within two quantization levels of the zero-level.

The function cleans up the silence parts of clean speech for very low level noise, and hereby enhances the perceptual quality of clean speech. The effect of the function becomes especially noticeable when the input originates from an A-law source, i.e. the input has passed through A-law encoding and decoding immediate prior to the speech coding algorithm. This is due to the amplification of sample values around zero (e.g. –1, 0, +1) to either –8 and +8 inherent in A-law. The amplification has the potential of transforming an inaudible "silence noise" into a clearly audible noise.

3.1.1.2 High-pass Filter

The input high-pass filter is identical to the input high-pass filter of G.729. It is a 2[nd] order pole-zero filter with a cut-off frequency of 140 Hz. It is given by the following transfer function: